

(12) UK Patent Application (19) GB (11) 2 094 592 A

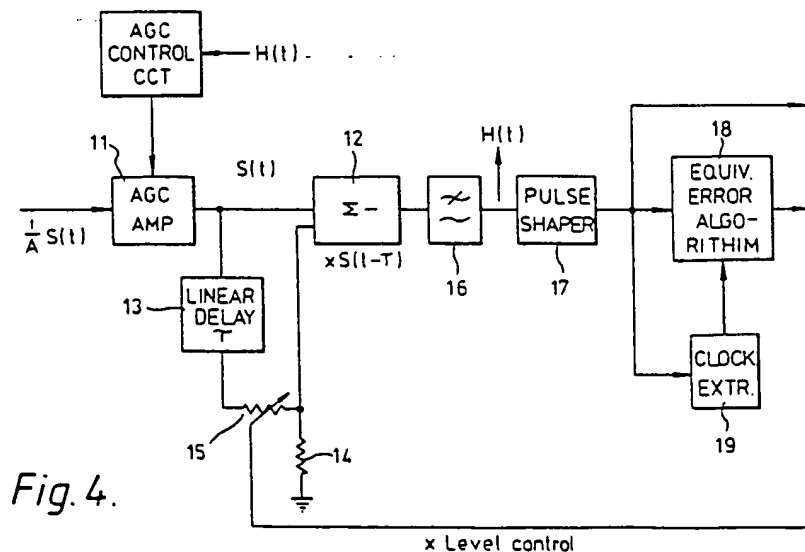
- (21) Application No 8104087
 (22) Date of filing 10 Feb 1981
 (43) Application published
 15 Sep 1982
 (51) INT CL¹
 H04B 3/14
 (52) Domestic classification
 H4R LEX
 (56) Documents cited
 GB 1526420
 GB 1138693
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 GB 1111289
 GB 1204593
 GB 0523434
 GB 1146100
 GB 0517516
 (58) Field of search
 H3U
 H3T
 H4R
 H4P

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(54) Adaptive equaliser

(57) In a digital system, especially where PCM coded speech is conveyed over twisted pair cables optimised for speech, a pulse is received distorted, the distortion being especially bad on the trailing edge of the pulse. This decays slowly enough to cause inter-symbol interference with the next pulse.

This is overcome by applying the signal via an amplifier (11) direct to one input of a subtractor (12) and via a delay 13 and variable attenuator (14-15) to the other input of the subtractor. The delay is such as to put the delayed and attenuated version at the bit time for the next pulse: hence the interference due to the slow decay of the trailing edge is eliminated.



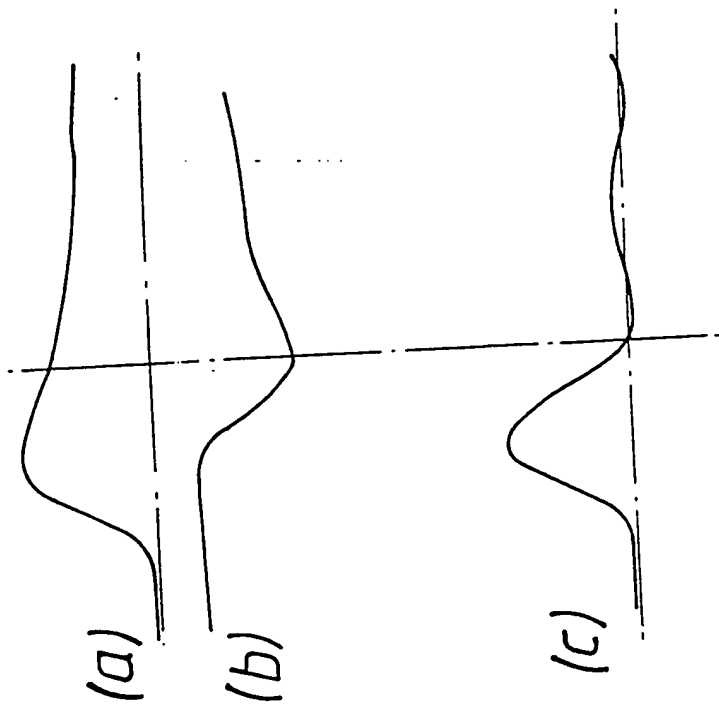


Fig.1.

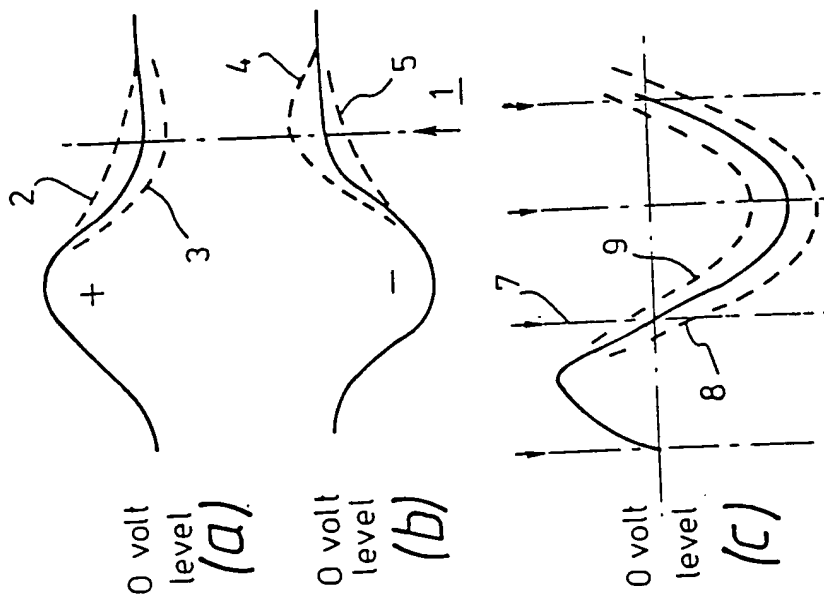


Fig.2.

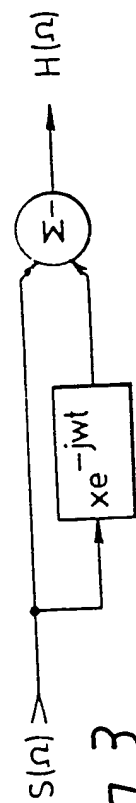


Fig.3.

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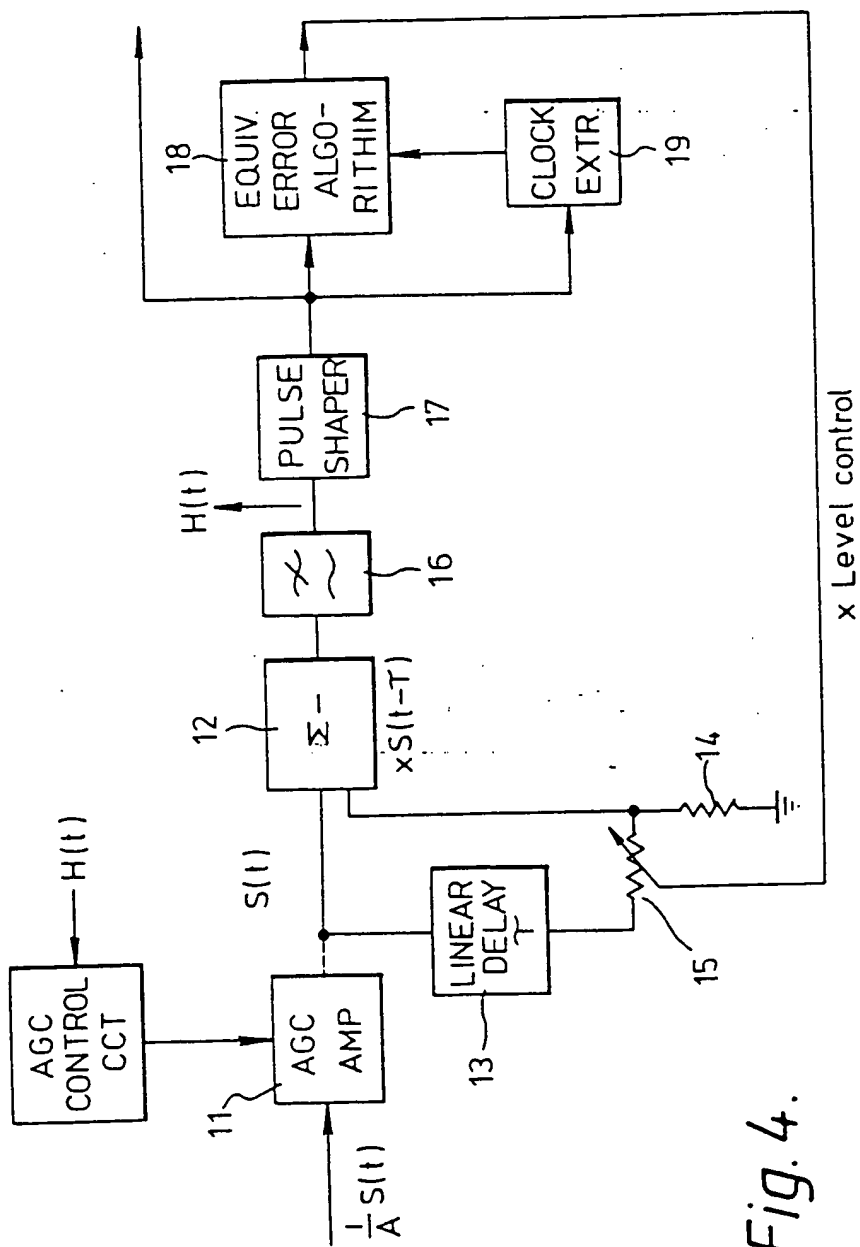


Fig. 4.

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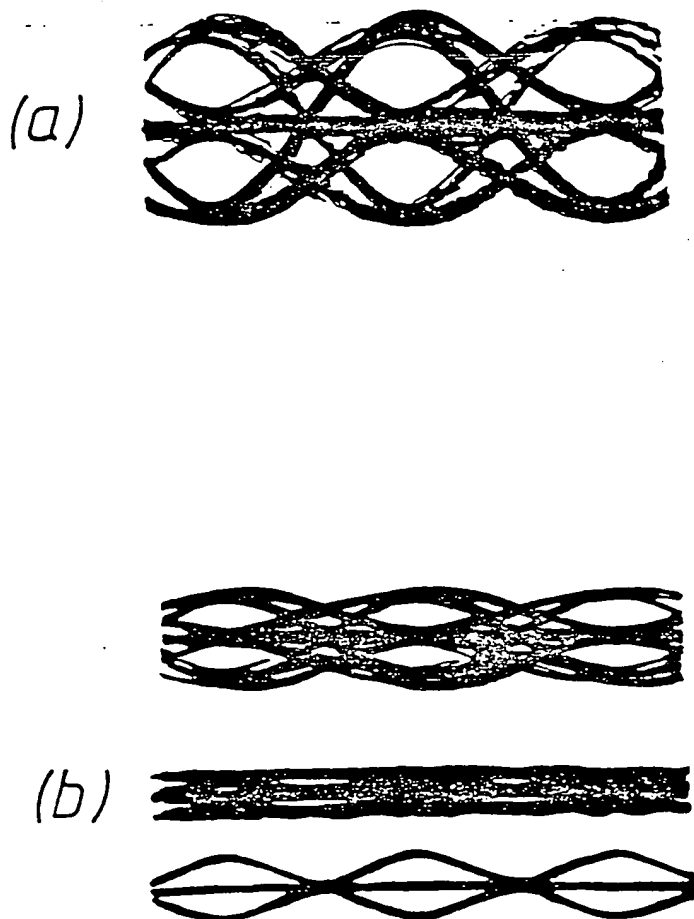


Fig. 5.

SPECIFICATION

Adaptive equaliser

5 This invention relates to an adaptive equaliser for use in a digital line transmission system.

Pulse transmission down telephone twisted pair cable is needed when speech is PCM encoded. For A-law PCM the basic information rate is 64K bits/sec; 10 in digital local area telephony the need for additional bits for synchronisation and signalling increases this to 80K bits/sec. For duplex operation on one pair of wires, a two-four wire hybrid technique can be used, in which case the line transmission rate remains at 15 80K bits/sec, or time separation can be used. In this case a number of PCM words, e.g. a whole frame, is assembled and sent out at a higher rate, usually of the order of three times the normal bit rate, with the wires conveying the bit bursts alternately in different 20 direction.

Since the relatively high bit rates have to be transmitted over twisted pair cables optimised for audio band transmission, the pulses would be considerably attenuated and distorted at the end of the 25 line. The waveform shown in Fig. 1a is an example showing the distortion to a 2 μ sec pulse after reception over 4Km of 0.5 mm transmission cable. The waveform is amplified and shows that the leading edge is fairly "well behaved", but the trailing edge 30 has a very slow decay, which can cause inter-symbol interference. For a continuous mode of transmission, where pulses of either polarity, or zero, may be transmitted consecutively, it is desirable to constrain the period of the pulses to within two periods, i.e. to 35 be zero at the centre of the adjacent pulses and at a maximum at its centre.

An object of the invention is to provide an equaliser in which the above desideratum is achieved.

According to the present invention there is provided an adaptive delay equaliser for use in the 40 reception of intelligence conveyed in pulse form, in which on reception the pulses are each converted and delayed by a preset time, and in which the delayed and inverted pulse is reduced in amplitude 45 by a variable amount and subtracted from the original pulse, the amount of the time delay and the amount of the amplitude reduction being such that the distortion present at the trailing edge of the pulse is substantially reduced or eliminated.

50 An embodiment of the invention will now be described with reference to the accompanying drawings, in which

Figs 1 and 2 are waveform diagrams used to explain the invention;

55 Fig 3 is a simple diagram explanatory of the principle of delay feedback equalisation.

Fig 4 is a simplified block diagram of an adaptive delay equaliser embodying the invention.

The basis of the equaliser will be clear from the 60 waveform of Fig. 1. As already mentioned, Fig. 1a shows one example of a pulse as received with distortions due to line conditions. Note especially that the most serious distortion is the slow delay of the pulse trailing edge. To minimise or eliminate this 65 distortion, a fraction of the main signal $S(t)$ is

delayed by a period τ , i.e. to give $xS(t-\tau)$, where $0 < x < 1$, and subtracted from the main signal. This fraction is shown inverted at Fig. 1b, and the result of the subtraction is shown at Fig. 1c. The period τ and the 70 fraction x are so chosen that the centre of the main pulse remains undisturbed and the point which corresponds to the centre of an adjacent pulse reaches zero.

Although the pulse duration is given as 2 μ sec it is 75 assumed that the pulse repetition rate is 4 μ sec. Thus the optimum duration for τ would lie between a half and one bit period, 2 to 4 τ sec in this case, dependent on the rising edge of the received pulse. Normally to optimise bandwidth we have found it 80 desirable for the delay to be one bit. The fraction of the delay signal required depends directly on the degree of distortion, and equals that portion of the main signal which remains 4 μ sec from the peak. The technique of pulse equalisation depends on the 85 degree of distortion and is independent of absolute amplitude.

The system whose principle is described above is made adaptive to compensate for varying degrees of distortion, i.e. line type and length, by devising an 90 algorithm which brings to zero the residue of any given pulse at the centre of the next pulse. This is illustrated for ternary transmission, i.e. Alternate Mark Inversion (AMI) code:

	Previous Pulse Polarity	Present Pulse Polarity	Correction
95	0	0	-
	+	-	-
	+	0	✓
100	-	+	-
	-	0	✓
	0	+	-
	0	-	-

105 Thus it will be seen that correction is only made from positive or negative pulses to zero. A simple data scrambling technique ensures that sufficient error correcting periods exist which can be averaged over a suitable time.

110 This correction algorithm is illustrated schematically in Fig. 2, in which Fig. 2a is for a positive pulse followed by zero and Fig. 2b is for a negative pulse followed by a zero. Hence the line 1 corresponds to the centre of the next bit period and the full line 115 curve is a correctly equalised pulse. The line 2 shows the state in which the pulse is under-equalised, so that x has to be increased. The line 3 represents the case in which the pulse is over-equalised so x has to be decreased. Similarly in Fig. 2b, the line 4 shows 120 the pulse over-equalised so x must be decreased while the line 5 shows the pulse under-equalised so x must be increased. Thus to summarise, for + to - or - to + one increases x , and for + to - or - to + one decreases x .

125 For two-level codes such as the Miller code or the dipulse code, the correction algorithm is simpler since if there is a transition at the bit boundary 7, Fig. 2c, it will be either early as at 8 if over-equalised or late as at 9 if under-equalised.

130 An equaliser embodying the above principles was

designed for use with a scrambled AML code at a bit rate of 256K bits/sec. and gave a good performance over a line length of 8 km with 0.5 mm copper distribution cable – see also the eye diagrams, Fig. 5.

- 5 The principle of delay feedback equalisation is illustrated schematically in Fig. 3, as defined by the following equation:

$$H(\omega) = S(\omega) \{1 - x e^{-j\omega\tau}\}, \text{ where } 0 < x < 1$$

- 10 We now turn to the simplified block diagram of Fig. 4 to give a more detailed description of an equaliser based on the above principles.

- 15 In the block diagram of Fig. 4, the incoming signal is defined as $\frac{1}{A} S(t)$, the $\frac{1}{A}$ indicating that it has been attenuated during its traversal of the line. This signal is applied to an AGC amplifier 11, whose output is the amplified signal $S(t)$; the AGC feature ensures that the output of this amplifier has a substantially constant amplitude. The signal $S(t)$ is applied directly to one input of a subtraction circuit 12, and indirectly thereto via a linear delay circuit 13 which introduces the delay τ . The connection from the delay circuit 13 to the other input of the subtraction circuit 12 is via an attenuator represented by a fixed resistor 14 and an adjustable resistor 15.

- 20 The output of the subtraction circuit passes via a low pass filter 16 whose output forms the output of the equaliser, to a pulse shaper 17. This output is applied to an equalisation error algorithm circuit 18 and to the clock extraction circuit 19. The latter circuit controls the circuit 18, whose output is indicative of the error condition of the received signal, and is used to adjust the attenuator, as shown by the connection to the variable resistor 15. Thus the value x is varied in the manner referred to above.

- By the technique described above it would also be possible to cancel out forward delay echoes due to bridged tap discontinuities along the transmission line. This would need at least one more tap delayed by 2τ and a separate x control. The error algorithm would then need correction for two consecutive zeroes, making use of an iterative approach.

- 45 We now refer briefly to Figs. 5a and b which are derived from photographs of eye diagrams for the delay feedback equaliser operating over 8Km if 0.5mm distribution cable. Data is sent in the basit mode at 252K bits/sa., using scrambled AML code, and the time base gives 1 μ s/div. Fig. 5a shows the eye at the receiver after automatic gains control and equalisation. Fig. 5b in the upper row shows the received eye after equalisation, in the middle row the received signal before equalisation, and in the bottom row the transmit signal.

CLAIMS

1. An adaptive delay equaliser for use in the reception of intelligence conveyed in pulse form, in which on reception the pulses are each inverted and delayed by a preset time, and in which the delayed and inverted pulse is reduced in amplitude by a variable amount and subtracted from the original pulse, the amount of the time delay and the amount of the

tially reduced or eliminated.

2. An equaliser as claimed in claim 1, in which the intelligence is conveyed in a ternary form so that the possible line conditions are a positive pulse, a zero state and a negative pulse, and in which the said delay and subtraction are only effected when a positive pulse is followed by a zero state or a negative pulse is followed by a zero state.

3. An adaptive equaliser for use in the reception of intelligence conveyed in pulse form, substantially as described with reference to the accompanying drawings.

New Claims or amendments to claims filed on 1 Dec. 1981.

Superseded claims all.

New claims: – 1 to 4

1. An adaptive delay equaliser for use in the reception of intelligence conveyed in pulse form, in which in reception the pulses are each delayed by a preset time, and in which the delayed pulse is reduced in amplitude by a variable amount and subtracted from the original pulse, the amount of the time delay and the amount of the amplitude reduction being such that the distortion present at the trailing edge of the pulse is substantially reduced or eliminated.

2. An equaliser as claimed in claim 1, in which the intelligence is conveyed in a ternary form so that the possible line conditions are a positive pulse, a zero state and a negative pulse, and in which the said delay and subtraction are only effected when a positive pulse is followed by a zero state or a negative pulse is followed by a zero state.

3. An adaptive equaliser for use in the reception of intelligence conveyed in pulse form, substantially as described with reference to the accompanying drawings.

4. An adaptive delay equaliser for use in the reception of intelligence conveyed in pulse form at a substantially constant pulse repetition frequency, in which the pulses as received have been subjected to distortion producing a slow decay of the trailing edge of a pulse, which slow decay causes interference (hereinafter referred to as inter-symbol interference) with the next received pulse, in which on reception the pulses are each applied via an amplifier to a first input of a subtraction circuit, in which the pulses as amplified by the amplifier are also applied via a delay circuit and an attenuator to a second input of the subtraction circuit, in which the delay introduced by the delay circuit is such that pulse as delayed is in synchronism with the next pulse to be received, and in which the attenuator is so adjusted as to set the amplitude of the pulse applied to the second input of the subtraction circuit to a level substantially equal to that of the interference due to the slow decay of the pulse's trailing edge, whereby the output from the subtraction circuit is a pulse sequence from which inter-symbol interference has been substantially eliminated.